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(54) **RECEIVER INTELLIGIBILITY
ENHANCEMENT SYSTEM**

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G10L 21/0208 (2013.01)

H04R 29/00 (2006.01)

H04R 3/00 (2006.01)

G10L 25/12 (2013.01)

(52) **U.S. Cl.**

CPC **G10L 21/0364** (2013.01); **G10L 21/0208** (2013.01); **H04R 3/005** (2013.01); **G10L 25/12** (2013.01)

(58) **Field of Classification Search**

None

See application file for complete search history.

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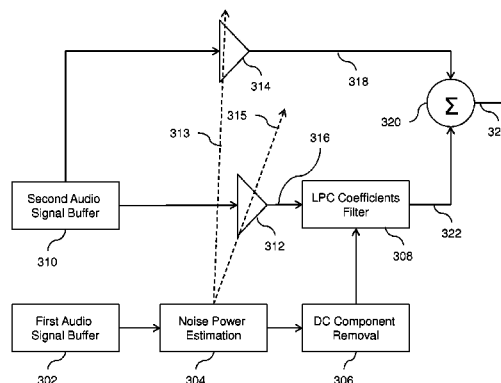
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(57) **ABSTRACT**

Embodiments of the invention provide a communication device and methods for enhancing audio signals. A first audio signal buffer and a second audio signal buffer are acquired. Thereafter, the second audio signal is processed based on the linear predictive coding coefficients and gains based on mean noise power of the first audio signal to generate an enhanced second audio signal.

14 Claims, 9 Drawing Sheets



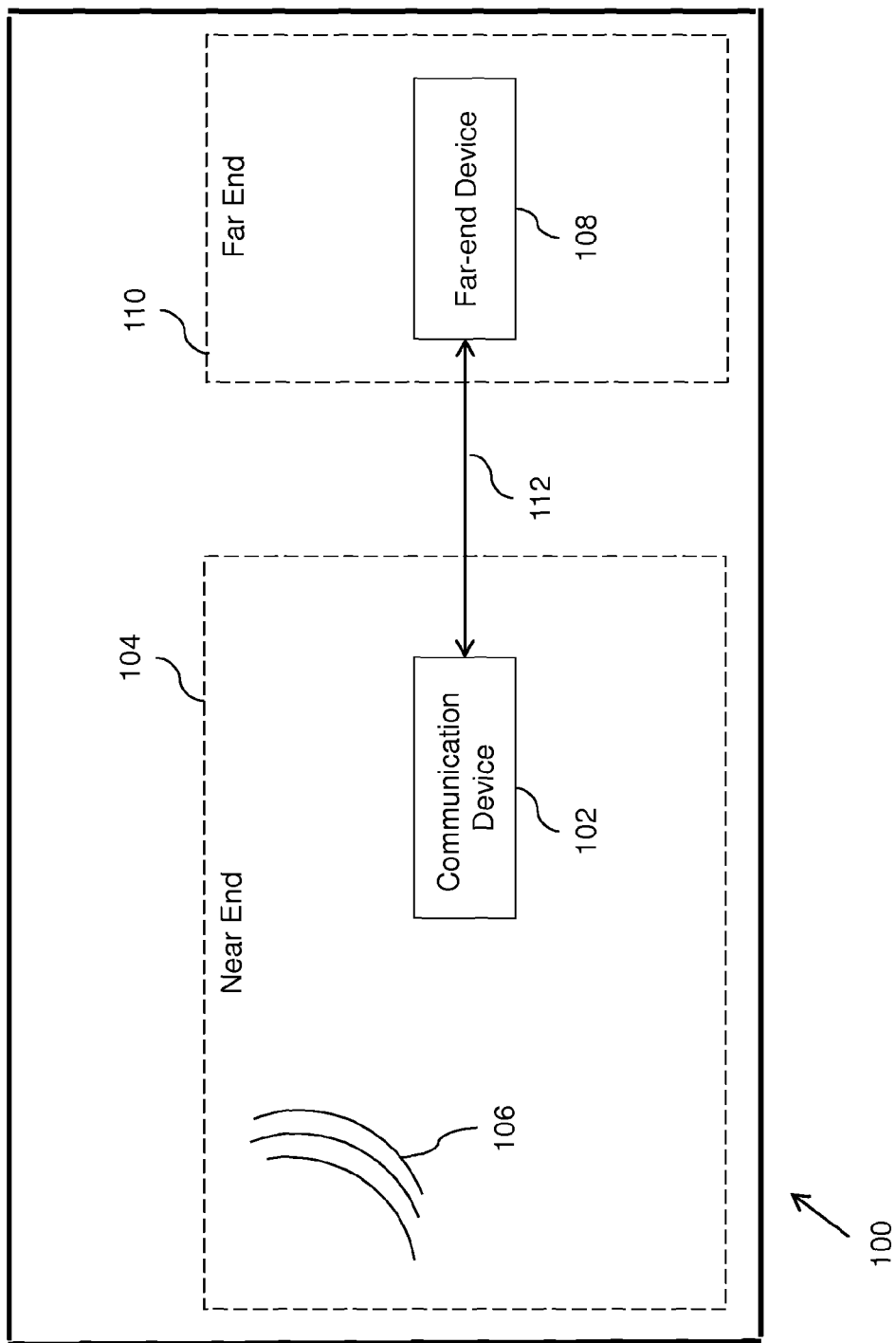


FIG. 1

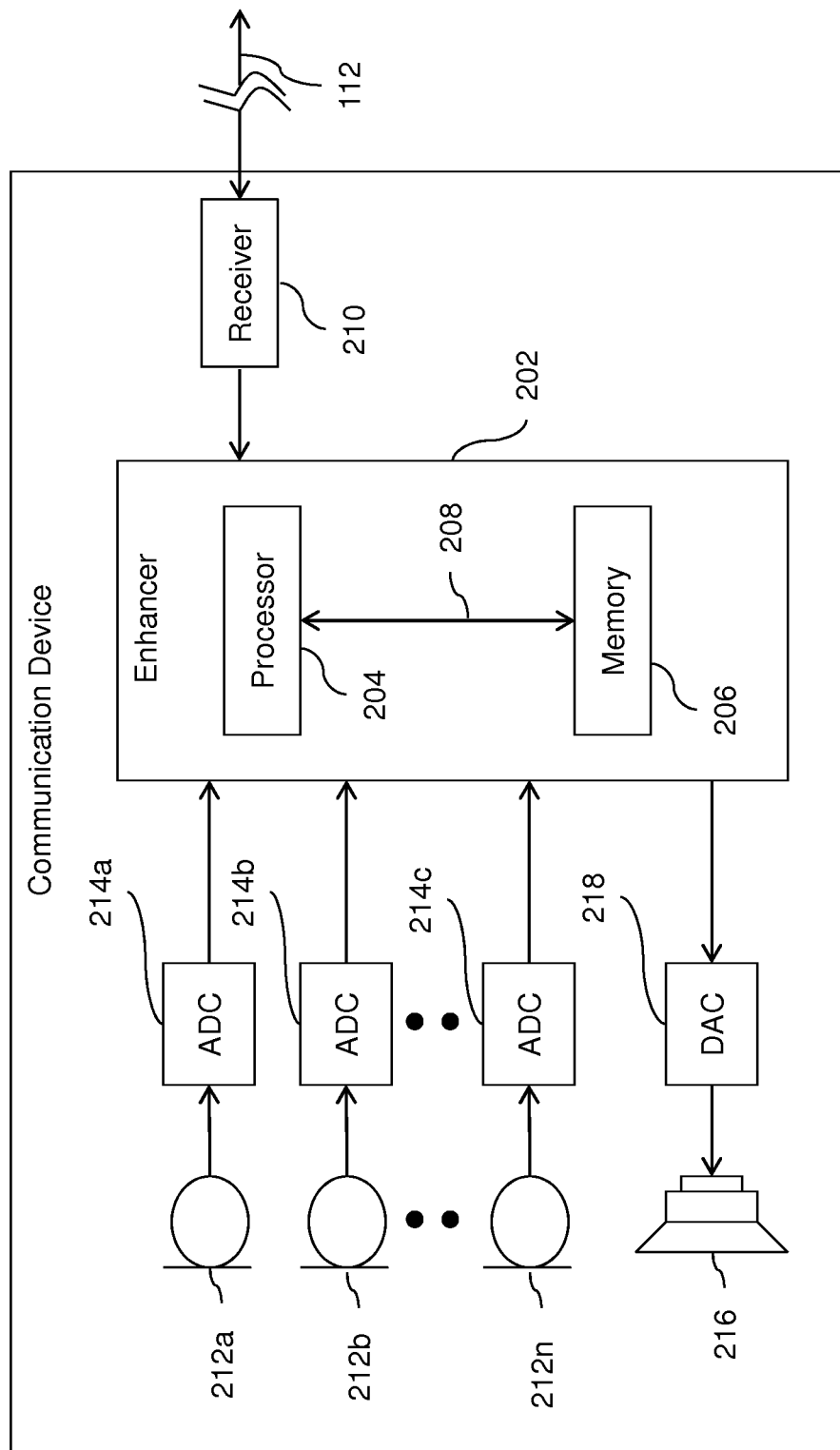


FIG. 2

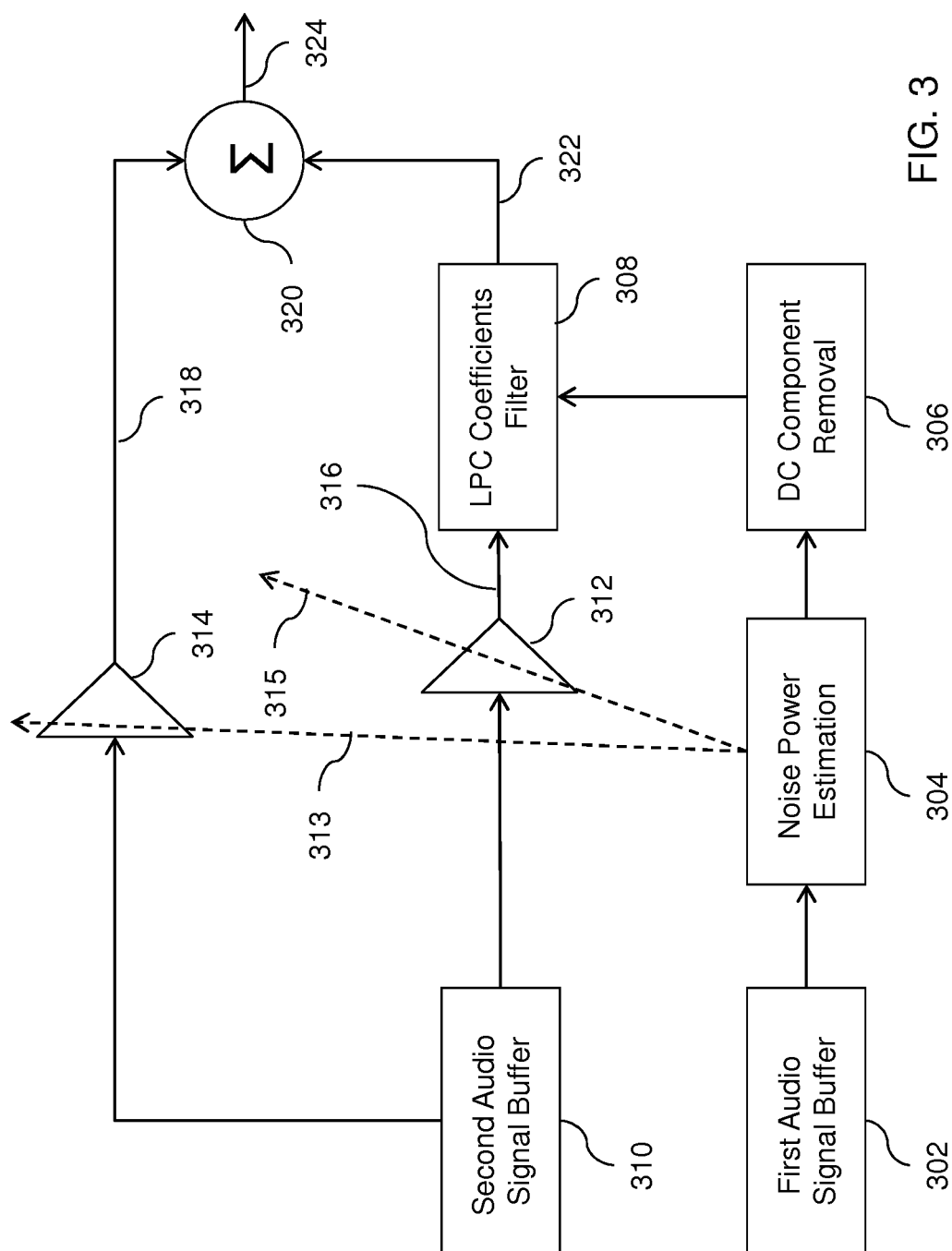


FIG. 3

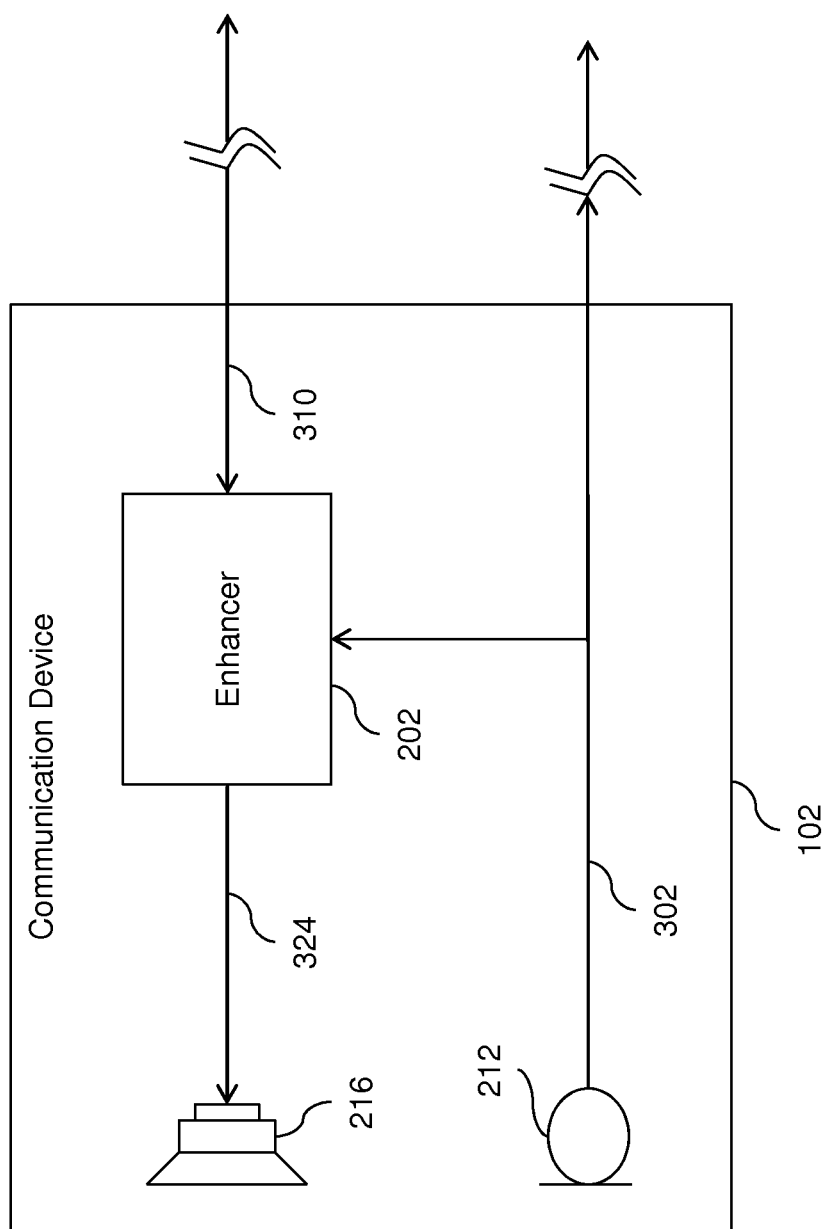


FIG. 4

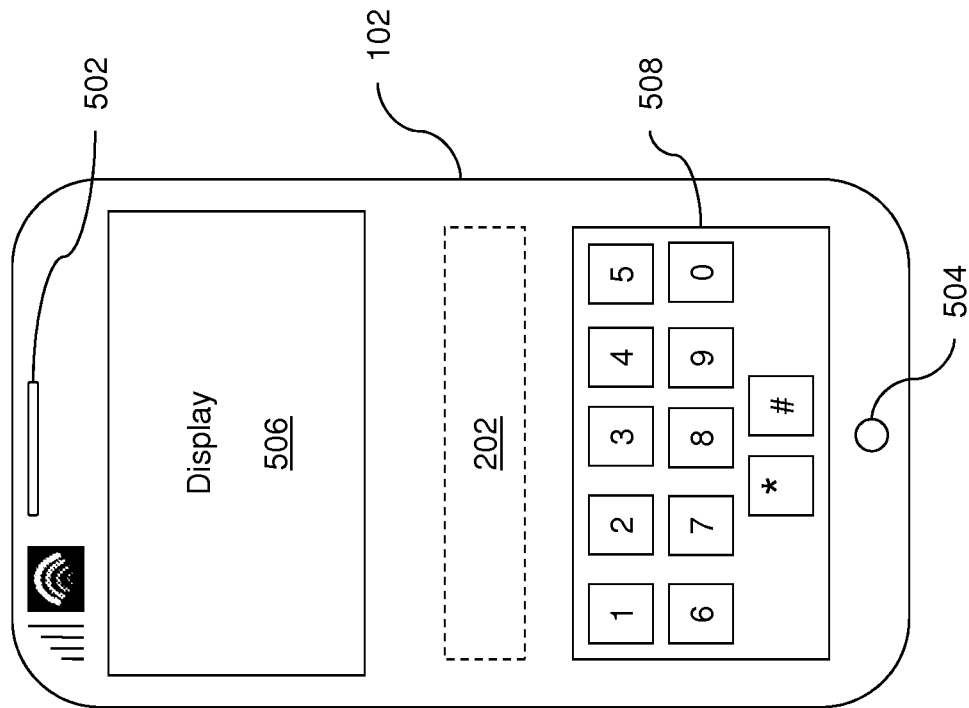


FIG. 5

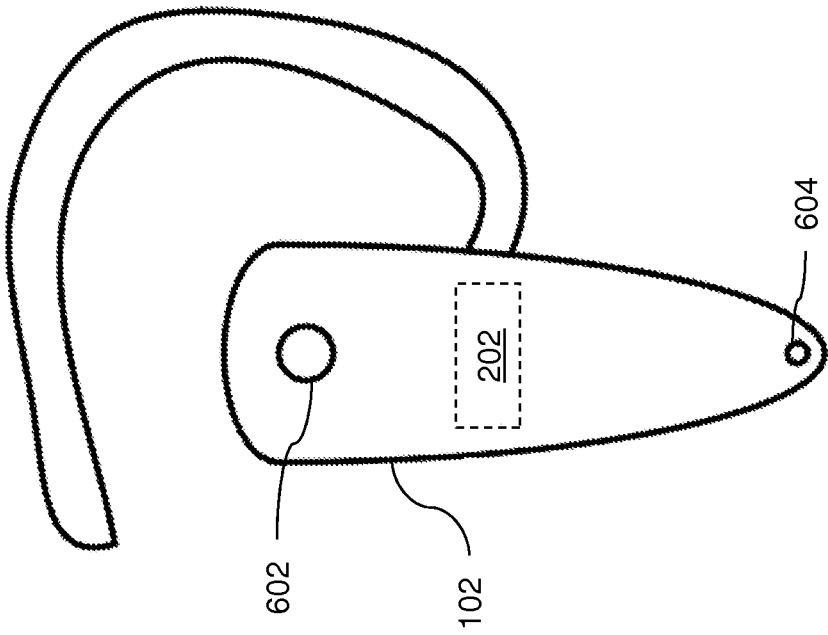


FIG. 6

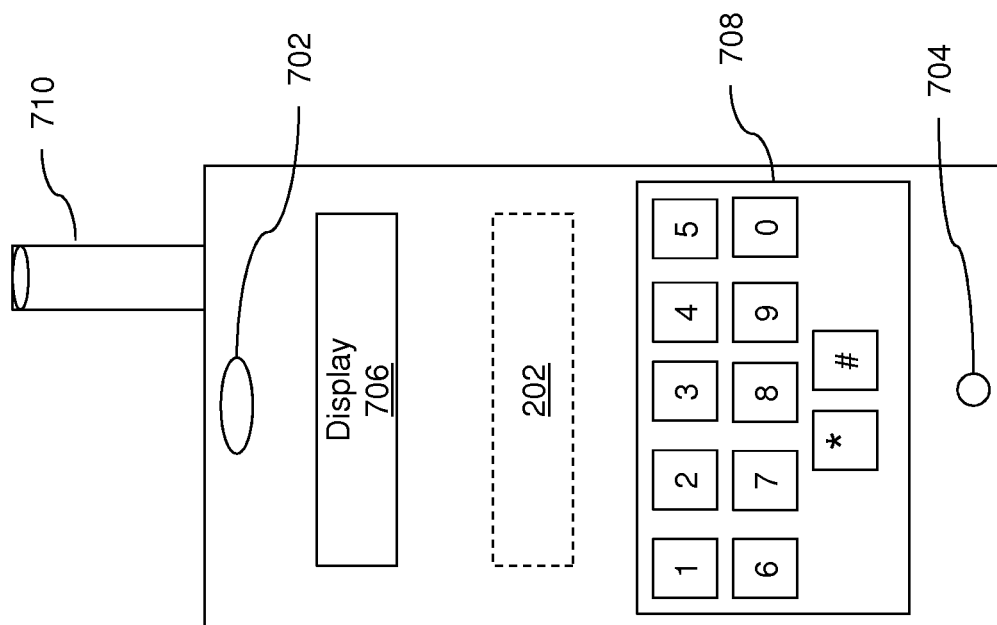


FIG. 7

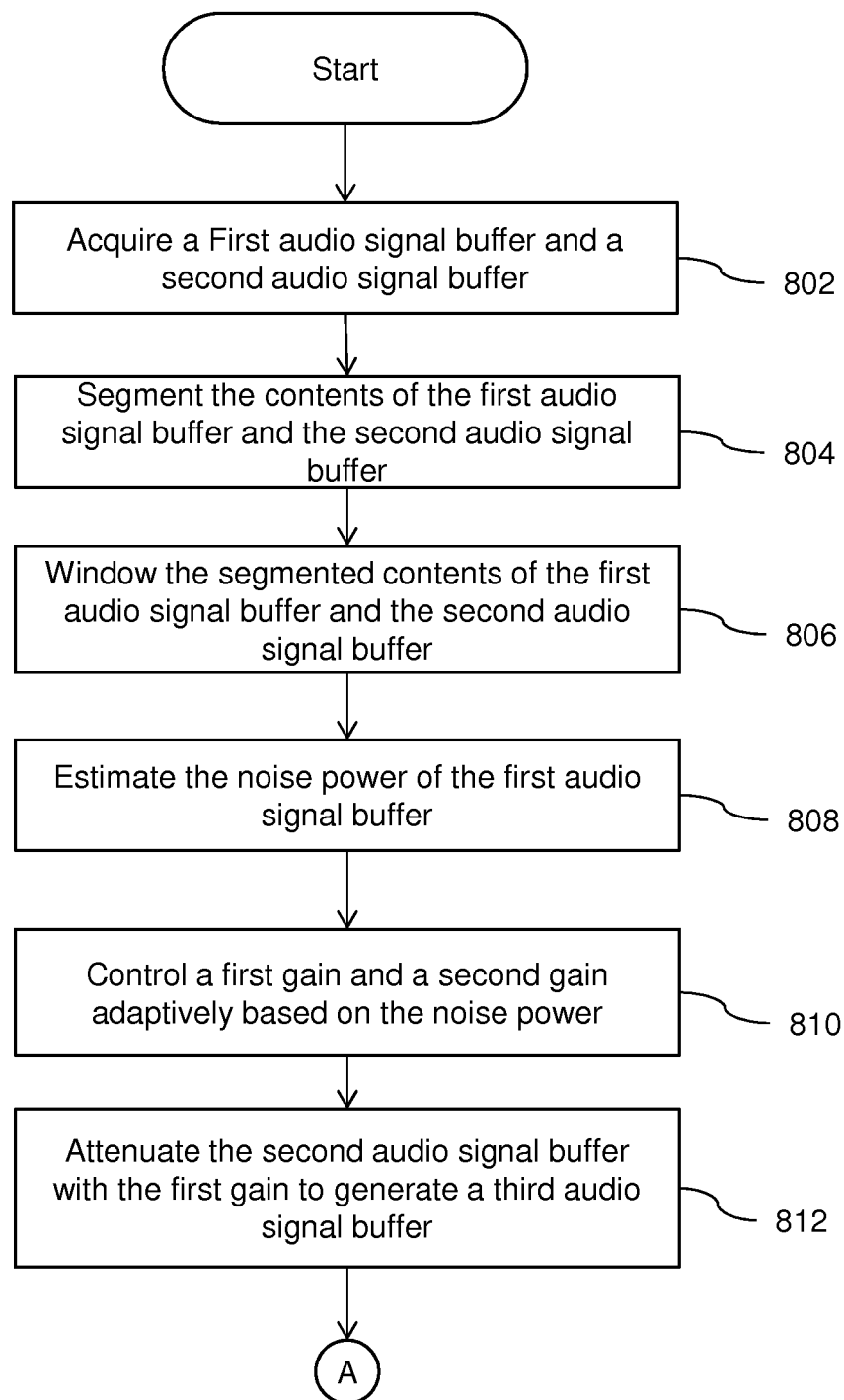


FIG. 8A

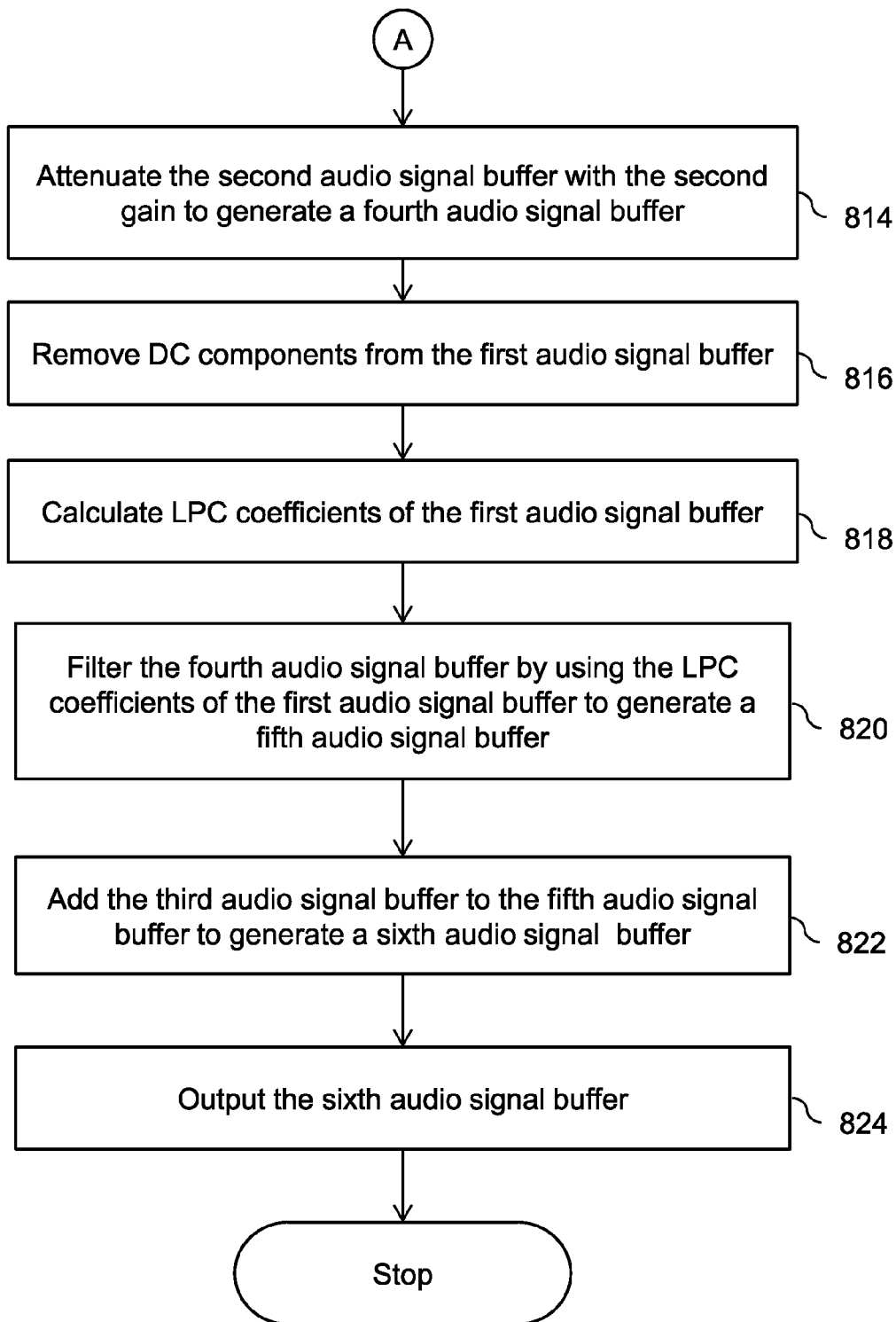


FIG. 8B

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RECEIVER INTELLIGIBILITY ENHANCEMENT SYSTEM

CROSS REFERENCE TO RELATED APPLICATIONS

This application claims the priority date and benefit of and is a continuation in part application of U.S. application Ser. No. 12/946,468 filed on Nov. 15, 2010 which is a continuation in part application of U.S. application Ser. No. 12/139,489 filed on Jun. 15, 2008, which claims priority from provisional application 60/944,180 filed on Jun. 15, 2007. The entire teachings and contents of the above related patent applications are incorporated herein by reference.

FIELD OF THE INVENTION

This invention relates to audio signal processing, and more specifically, the invention relates to systems and methods for enhancing receiver intelligibility.

BACKGROUND

Speech intelligibility is usually expressed as a percentage of words, sentences or phonemes correctly identified by a listener or a group of listeners. It is an important measure of the effectiveness or adequacy of a communication system or of the ability of people to communicate effectively in noisy environments. Quality is a subjective measure, which reflects on individual preferences of listeners. The two measures are not correlated. In fact, it is well known that intelligibility can be improved if one is willing to sacrifice quality. It is also well known that improving the quality of a signal does not necessarily elevate its intelligibility. On the contrary, quality improvement is usually associated with loss of intelligibility relative to that of the signal. This is due to distortion that the signal undergoes in the process of enhancing it.

Communication devices such as mobile phones, headsets, telephones and so forth may be used in vehicles or in other areas where there is often a high level of background noise. A high level of local background noise can make it difficult for a user of the communication device to understand the speech being received from the receiving side in the communication network. The ability of the user to effectively understand the speech received from the receiver side is obviously essential and is referred to as the intelligibility of the received speech.

In the past, the most common solution to overcome the background noise was to increase the volume at which the speakers of communication device output speech. One problem with this solution is that the maximum output sound level that a phone's speaker can generate is limited. Due to the need to produce cost-competitive cell phones, companies often use low-cost speakers with limited power handling capabilities. The maximum sound level such phone speakers generate is often insufficient due to high local background noise.

Attempts to overcome the local background noise by simply increasing the volume of the speaker output can also result in overloading the speaker. Overloading the loudspeaker introduces distortion to the speaker output and further decreases the intelligibility of the outputted speech. A technology that increases the intelligibility of speech received irrespective of the local background noise level is needed.

Several attempts to improve the intelligibility in communication devices are known in the related art. The requirements of an intelligent system cover naturalness of the enhanced signal, short signal delay and computational simplicity.

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During the past two decades, Linear Predictive Coding (LPC) has become one of the most prevalent techniques for speech analysis. In fact, this technique is the basis of all the sophisticated algorithms that are used for estimating speech parameters, for example, pitch, formants, spectra, vocal tract and low bit representations of speech. The basic principle of linear prediction states that speech can be modeled as the output of a linear time-varying system excited by either periodic pulses or random noise. The most general predictor form in linear prediction is the Auto Regressive Moving Average (ARMA) model where a speech sample of 's(n)' is predicted from 'p' past predicted speech samples s(n-1), . . . , s(n-p) with the addition of an excitation signal u(n) according to the following equation 1:

$$s(n) = \sum_{k=1}^p a_k s(n-k) + G \sum_{l=0}^q b_l u(n-l) \quad \text{Equation 1}$$

where G is the gain factor for the input speech and a.sub.k and b.sub.1 are filter coefficients. The related transfer function H(z) is given by following equation 2:

$$H(z) = S(z)/U(z) \quad \text{Equation 2}$$

For an all-pole or Autoregressive (AR) model, the transfer function becomes as the following equation 3:

$$H(z) = 1 / (1 - \sum_{k=1}^p a_k z^{-k}) = 1/A(z) \quad \text{Equation 3}$$

Estimation of LPC

Two widely used methods for estimating the LP coefficients exist: autocorrelation method and covariance method. Both methods choose the LP coefficients a.sub.k in such a way that the residual energy is minimized. The classical least squares technique is used for this purpose. Among different variations of LP, the autocorrelation method of linear prediction is the most popular. In this method, a predictor (an FIR of order m) is determined by minimizing the square of the prediction error, the residual, over an infinite time interval. Popularity of the conventional autocorrelation method of LP is explained by its ability to compute a stable all-pole model for the speech spectrum, with a reasonable computational load, which is accurate enough for most applications when presented by a few parameters. The performance of LP in modeling of the speech spectrum can be explained by the autocorrelation function of the all-pole filter, which matches exactly the autocorrelation of the input signal between 0 and m when the prediction order equals m. The energy in the residual signal is minimized. The residual energy is given by the following equation 4:

$$E = \sum_{n=-\infty}^{\infty} e^2(n) = \sum_{n=-\infty}^{\infty} [s_n(n) - \sum_{k=1}^m a_k s_n(n-k)]^2 \quad \text{Equation 4}$$

The covariance method is very similar to the autocorrelation method. The basic difference is the length of the analysis window. The covariance method windows the error signals instead of the original signal. The energy E of the windowed error signal is given by following equation 5:

$$E = \sum_{n=-\infty}^{\infty} e^2(n) = \sum_{n=-\infty}^{\infty} e^2(n) w(n) \quad \text{Equation 5}$$

Comparing autocorrelation method and covariance method, the covariance method is quite general and can be used with no restrictions. The only problem is that of stability of the resulting filter, which is not a severe problem generally. In the autocorrelation method, on the other hand, the filter is guaranteed to be stable, but the problems of parameter accuracy can arise because of the necessity of windowing the time signal. This is usually a problem if the signal is a portion of an impulse response.

Usually in environments with significant local background noise, the signal received from the receiving side becomes unintelligible due to a phenomenon called masking. There are

several kinds of masking, including but not limited to, auditory masking, temporal masking, simultaneous masking and so forth.

Auditory masking is a phenomenon when one sound is affected by the presence of another sound. Temporal masking is a phenomenon when a sudden sound makes other sounds inaudible. Simultaneous masking is the inability of hearing a sound in presence of other sound whose frequency component is very close to desired sound's frequency component.

In light of the above discussion, techniques are desirable for enhancing receiver intelligibility.

SUMMARY

The present invention provides a communication device and method for enhancing audio signals. The communication device may monitor the local background noise in the environment and enhances the received communication signal in order to make the communication more relaxed. By monitoring the ambient or environmental noise in the location in which the communication device is operating and applying receiver intelligibility enhancement processing at the appropriate time, it is possible to significantly improve the intelligibility of the received communication signal.

In one aspect of the invention, the noise in the background in which the communication device is operating is monitored and analyzed.

In another aspect of the invention, the signals from a far-end are modified based on the characteristics of the background noise at near end.

In another aspect of the invention, Linear Predictive Coding (LPC) coefficients of a first audio signal buffer acquired from a near-end are used to filter a second audio signal buffer acquired from a far-end to generate an intelligibility enhanced signal.

BRIEF DESCRIPTION OF THE DRAWINGS

Having thus described the invention in general terms, reference will now be made to the accompanying drawings, which are not necessarily drawn to scale, and wherein:

FIG. 1 illustrates an environment where various embodiments of the invention function;

FIG. 2 illustrates a block diagram of a communication device for enhancing audio signals, in accordance with an embodiment of the invention;

FIG. 3 is a flow diagram illustrating processing of audio signals, in accordance with an embodiment of the invention;

FIG. 4 illustrates acquiring and outputting of audio signals by the communication device, in accordance with an embodiment of the invention;

FIG. 5 illustrates the communication device as a mobile phone, in accordance with an embodiment of the invention;

FIG. 6 illustrates the communication device as a headset, in accordance with an embodiment of the invention;

FIG. 7 illustrates the communication device as a cordless phone, in accordance with an embodiment of the invention;

FIG. 8A is a flowchart illustrating enhancing of audio signal, in accordance with an embodiment of the invention; and

FIG. 8B is a flowchart in accordance with an embodiment of the invention.

DETAILED DESCRIPTION OF THE INVENTION

The following detailed description is directed to certain specific embodiments of the invention.

However, the invention can be embodied in a multitude of different ways as defined and covered by the claims and their equivalents. In this description, reference is made to the drawings wherein like parts are designated with like numerals throughout. Unless otherwise noted in this specification or in the claims, all of the terms used in the specification and the claims will have the meanings normally ascribed to these terms by workers in the art.

The present invention provides a novel and unique technique to improve the intelligibility in noisy environments experienced in communication devices such as a cellular telephone, wireless telephone, cordless telephone, and so forth. While the present invention has applicability to at least these types of communications devices, the principles of the present invention are particularly applicable to all types of communications devices, as well as other devices that process speech in noisy environments such as voice recorders, dictation systems, voice command and control systems, and the like. For simplicity, the following description may employ the terms "telephone" or "cellular telephone" as an umbrella term to describe the embodiments of the present invention, but those skilled in the art will appreciate that the use of such term is not to be considered limiting to the scope of the invention, which is set forth by the claims appearing at the end of this description.

FIG. 1 illustrates an environment 100 where various embodiments of the invention function. A communication device 102 may communicate with a far-end device 108 through a communication channel 112. Examples of communication device 102 and far-end device 108 include, but are not limited to, a mobile phone, a telephone, a cordless phone, a Bluetooth headset, a computer, a dictation system, voice recorders and other devices capable of communication. Communication channel 112 may be for example, a wireless channel, a radio channel, a wired channel and so forth. Communication device 102 and far-end device 108 communicate by exchanging signals over communication channel 112. Far-end device 108 may be located at a far end 110 from communication device 102, while communication device 102 may be located at a near end 104. Far end 110 may be location that is distant from near end 104 of communication device 102. For example, near end 104 may be a restaurant having local background noise 106 and far end 110 may be a home or office. Background noise 106 may be due to talking of other people, machines or devices used inside or near the restaurant.

Generally in conventional devices the signals received from far-end device 108 and outputted through an earpiece of the communication device 102 may not sound clear because of the background noise 106. The present invention provides techniques to generate and output clear and enhanced signals from the earpiece of communication device 102.

FIG. 2 illustrates a block diagram of communication device 102 for enhancing audio signals, in accordance with an embodiment of the invention. Communication device 102 may include multiple microphones 212a-n for acquiring audio signals. The audio signals acquired by microphones 212a-n may be analog and can be converted to digital audio signals by Analog-To-Digital (ADC) convertors 214a-n connected to microphones 212a-n. Microphones 212a-n may acquire audio signals from near end 104 of communication device 102. Therefore, the audio signals acquired by microphones 212a-n may include background noise. Although, multiple microphones 212a-n are shown, a person skilled in the art will appreciate that the present invention can function with a single microphone implemented in communication device 102.

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A Digital-To-Analog (DAC) convertor **218** connected to an earpiece **216** may convert digital audio signals to analog audio signals that may then be outputted by earpiece **216**. Further, communication device **102** includes a receiver **210** that receives signals from a far-end device on communication channel **2**. An enhancer **202** processes the signals received from microphones **212a-n** and receiver **210** to enhance the signal received from receiver **210**. Further, the enhanced signal is outputted from earpiece **216**. Enhancer **202** may include a processor **204** and a memory **206**. Processor **204** can be a general purpose fixed point or floating point Digital Signal Processor (DSP), or a specialized DSP (fixed point or floating point). Examples of processor **204** include, but are not limited to, processor Texas Instruments (TI) TMS320VC5510, TMS320VC6713, TMS320VC6416; Analog Devices (ADI) BlackFinn (BF) 531, BF532, 533; Cambridge Silicon Radio (CSR) Blue Core 5 Multi-media (BC5-MM) or Blue Core 7 Multi-media BC7-MM and so forth. Memory **206** can be for example, a Random Access Memory (RAM), SRAM (Static Random Access Memory), a Read Only Memory (ROM), a solid state memory, a computer readable media and so forth. Further, memory **206** may be implemented inside or outside communication device **102**. Memory **206** may include instructions that can be executed by processor **204**. Further, memory **206** may store data that may be used by processor **204**. Processor **204** and memory **206** may communicate for data transfer through system bus **208**.

FIG. 3 is a flow diagram illustrating processing of audio signals, in accordance with an embodiment of the invention. Background noise **106** acquired by microphones **212a-n** may be converted to digital first audio signal buffer **302**. Similarly, audio signals received from far end **110** may be processed as second audio signal buffer **310**. The audio signals received from far end **110** can be speech signals. In an embodiment of the invention, background noise **106** and audio signals received from far end **110** may be stored as digital first audio signal buffer **302** and second audio signal buffer **310** respectively in memory **206** for processing. Further, the contents of first audio signal buffer **302** and second audio signal buffer **310** may be segmented and windowed for processing. In an embodiment of the invention, the segmentation is done by using a Hanning window. However people skilled in the art can appreciate the fact that the other windowing schemes, such as Hamming window, Blackman-Harris window, trapezoidal window and so forth, can also be used.

At block **304**, noise power of first audio signal buffer **302** may be calculated. For example, the noise power can be calculated as shown by pseudo program instructions and equation:

| | |
|---|------------|
| Noise Power = 0 | Equation 6 |
| Loop i = 1 to P | |
| Noise Power = Noise Power + input[i] ² | |
| End Loop | |

where 'i' is an index, 'P' is the number of samples in each frame of first audio signal buffer **302**. For example, there can be 160 samples in each frame for a narrowband communication system. In equation (1), 'input[i]' represents first audio signal buffer **302**. The result of the above mentioned instructions is the 'Noisepower' of first audio signal buffer **302**. In an embodiment of the invention, the above mentioned instructions may be stored in memory **206**.

Second audio signal buffer **310** is attenuated at a block **314** by a first gain **313** to generate a third audio signal buffer **318**.

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First gain **313** may be within a first predefined range. For example the first predefined range may be from 10% to 30%. Moreover, second audio signal buffer **310** is attenuated at a block **312** by a second gain **315** to generate a fourth audio signal buffer **316**. Second gain **315** may be within a second predefined range. For example the second predefined range may be from 70% to 90%. Therefore, the sum of first gain **313** and second gain **315** may equal 100%. The values of gain can be controlled adaptively based on the noise power.

The mean of the noise power (MeanNoisePower) can be calculated by using equation (2):

| | |
|---------------------------------|------------|
| MeanNoisePower=MeanNoisePower/P | Equation 7 |
|---------------------------------|------------|

Further, Direct Current (DC) components can be removed from first audio signal buffer **302** as shown by pseudo program instructions and equation:

| | |
|---|------------|
| Loop i = 1 to P | Equation 8 |
| Input[i] = input[i] ² - MeanNoisePower | |
| End Loop | |

At block **308**, fourth audio signal buffer **316** may be filtered by using Linear Prediction Coding (LPC) coefficients to generate a fifth audio signal buffer **322**. The LPC coefficients are calculated based on the components of first audio signal buffer **302** after the removal of DC components. In an embodiment of the invention, the LPC coefficients may be calculated using Durbin-Levinson method. However, people skilled in the art will appreciate that other techniques such as covariance method, autocorrelation method or other methods may be used to calculate the LPC coefficients. Thereafter, fifth audio signal buffer **322** is added to third audio signal buffer **318** at block **320** to generate a sixth audio signal buffer **324**. Sixth audio signal buffer **324** is an enhanced audio signal that may be converted from digital to analog and outputted from earpiece **216** of communication device **102**. In an embodiment of the invention first audio signal buffer **302**, the second audio signal buffer **310**, third audio signal buffer **318**, fourth audio signal buffer **316**, fifth audio signal buffer **322**, and sixth audio signal buffer **324** may be stored in memory **206** for processing by processor **204**.

FIG. 4 illustrates acquiring and outputting of audio signals by communication device **102**, in accordance with an embodiment of the invention. As shown, first audio signal buffer **302** is acquired from microphone **212** and second audio signal buffer **310** is received from far-end device **108**. Communication device **102** transmits signals to far-end device **108** based on first audio signal buffer **302**.

First audio signal buffer **302** and second audio signal buffer **310** are processed by enhancer **202** to generate sixth audio signal buffer **324**. Sixth audio signal buffer **324** may be converted from digital to analog and outputted from earpiece **216** of communication device **102**. Sixth audio signal buffer **324** is an enhanced form of second audio signal buffer **310** that sounds clear to the user of communication device **102** even in presence of background noise **106**.

FIG. 5 illustrates communication device **102** as a mobile phone, in accordance with an embodiment of the invention. As shown, communication device **102** may include an earpiece **502**, a microphone **504**, a display **506**, a keypad **508**, and enhancer **202**. Further, mobile phone may communicate to another device through a mobile network. Microphone **504** acquires first audio signal buffer **302** and second audio signal buffer **310** is received from the other device on the mobile network. Although a single microphone **504** is shown, a person skilled in the art will appreciate that the mobile phone

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may include multiple microphones. Enhancer **202** processes first audio signal buffer **302** and second audio signal buffer **310** to generate an enhanced signal that is outputted from earpiece **502**. In an embodiment of the invention, communication device **102** may include a switch (not shown) to activate and/or deactivate enhancer **202**. Therefore, once enhancer **202** is deactivated, first audio signal buffer **302** and second audio signal buffer **310** are not processed and signal received from a far end device is outputted from earpiece **502**.

FIG. 6 illustrates communication device **102** as a headset, in accordance with an embodiment of the invention. Communication device **102** may be a Bluetooth headset that can be coupled with a device such as a mobile phone. As shown, the headset may include an earpiece **602**, a microphone **604** and enhancer **202**. Microphone **604** acquires first audio signal buffer **302** and second audio signal buffer **310** is received from the other device on radio or wireless channel. Although a single microphone **604** is shown, a person skilled in the art will appreciate that the mobile phone may include multiple microphones. Enhancer **202** processes first audio signal buffer **302** and second audio signal buffer **310** to generate an enhanced signal that is outputted from earpiece **602**. In an embodiment of the invention, communication device **102** may include a switch (not shown) to activate and/or deactivate enhancer **202**. Therefore, once enhancer **202** is deactivated, first audio signal buffer **302** and second audio signal buffer **310** are not processed and signal received from a far end device is outputted from earpiece **602**.

FIG. 7 illustrates communication device **102** as a cordless phone, in accordance with an embodiment of the invention. As shown, the cordless may include an earpiece **702**, a microphone **704**, a display **706**, a keypad **708**, an antenna **710** and enhancer **202**. The cordless phone may communicate with a far end device through a docking station (not shown) by using antenna **710**. Microphone **704** acquires first audio signal buffer **302** and second audio signal buffer **310** is received from the other device on radio or wireless channel. Although a single microphone **704** is shown, a person skilled in the art will appreciate that the mobile phone may include multiple microphones. Enhancer **202** processes first audio signal buffer **302** and second audio signal buffer **310** to generate an enhanced signal that is outputted from earpiece **702**. In an embodiment of the invention, earpiece **702** may include a loudspeaker.

In an embodiment of the invention, communication device **102** may include a switch (not shown) to activate and/or deactivate enhancer **202**. Therefore, once enhancer **202** is deactivated, first audio signal buffer **302** and second audio signal buffer **310** are not processed and signal received from a far end device is outputted from earpiece **702**.

FIG. 8 is a flowchart illustrating enhancing of audio signal, in accordance with an embodiment of the invention. Communication device **102** may communicate with far-end device **108** over communication channel **2**. However, communication device **102** may be present at a location having background noise. Therefore, the signals received from far-end device **108** may be required to be enhanced to make them clear and audible. At step **802**, first audio signal buffer **302** is acquired from microphones **212a-n** and second audio signal buffer **310** is acquired from far-end device **108**. Thereafter, at step **804**, the contents of first audio signal buffer **302** and second audio signal buffer **310** are segmented. At step **806**, the segmented contents of first audio signal buffer **302** and second audio signal buffer **310** are windowed. In an embodiment of the invention, the segmented contents are windowed based on Hanning window. Thereafter, at step **808**, noise power of first audio signal buffer **302** is estimated. Further, a

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mean noise power may be estimated at step **808**. Subsequently, at step **810**, first gain **313** and second gain **315** are generated and controlled. First gain **313** and second gain **315** are generated based on the noise power of first audio signal buffer **302**. Moreover, first gain **313** and second gain **315** can be controlled adaptively based on the noise power. In an embodiment of the invention, first gain **313** is within a first predefined range and second gain **315** is within a second predefined range. Further, the sum of first gain **313** and second gain **315** equals 100%.

Thereafter, at step **812**, second audio signal buffer **310** is attenuated by first gain **313** to generate third audio signal buffer **318**. Further, at step **814**, second audio signal buffer **310** is attenuated second gain **315** to generate fourth audio signal buffer **316**. In an embodiment of the invention, steps **812** and **814** may be performed simultaneously. At step **816**, DC components are removed from first audio signal buffer **302**. Thereafter, LPC coefficients of the first audio signal buffer **302** are calculated. At step **820**, fourth audio signal buffer **316** is filtered based on the LPC coefficients to generate fifth audio signal buffer **322**. Subsequently, fifth audio signal buffer **322** is added to third audio signal buffer **318** to generate sixth audio signal buffer **324**. Sixth audio signal buffer **324** may be converted from digital to analog and outputted from earpiece **216** of communication device **102**.

In one embodiment of the present invention, a communication device for generating enhanced audio signals and a method thereof is disclosed. The communication device comprises a first receiver receiving noise signals from a near-end location of the system; a second receiver configured to receive audio signals from far-end communication devices; a processor configured to process the noise signals and audio signals to enhance the audio signals, wherein firstly a mean noise power of the noise signal is estimated and secondly a linear predictive coding coefficients and gains are calculated based on the mean noise power to generate enhanced audio signals; and an earpiece to output the enhanced audio signals.

In one embodiment of the present invention, the gains comprise a first gain calculated within a first predefined range and a second gain calculated within a second predefined range, wherein the first predefined range is between 10% to 30% and the second predefined range is between 70% to 90%. Further, the sum of the first gain and the second gain is 100%.

The processor is configured to: attenuate the audio signals with the first gain to generate a third audio signal; attenuate the audio signals with the second gain to generate a fourth audio signal; filter the fourth audio signal buffer by using LPC coefficients of the noise signals to generate a fifth audio signal; and add the fourth audio signal buffer to the fifth audio signal buffer to generate the enhanced audio signals outputted by the earpiece.

A memory is also configured to store the noise signals, audio signals, third audio signal, fourth audio signal, fifth audio signal and the enhanced audio signals. The memory further stores one or more program instructions executable by the processor to process and enhance the audio signals. The processor is further configured to remove direct current components from the noise signals based on the mean noise power.

In one embodiment of the present invention, the processor is further configured to window the segmented contents of the noise signals and the audio signals by using Hanning window.

The audio signals being received by the second receiver comprise speech signals and are received through a communication channel. Said communication channel is a wireless communication channel.

In one embodiment of the present invention, the mean noise power is estimated based on a plurality of samples in a plurality of frames of the noise signals.

This written description uses examples to disclose the invention, including the best mode, and also to enable any person skilled in the art to practice the invention, including making and using any devices or systems and performing any incorporated methods. The patentable scope the invention is defined in the claims, and may include other examples that occur to those skilled in the art. Such other examples are intended to be within the scope of the claims if they have structural elements that do not differ from the literal language of the claims, or if they include equivalent structural elements with insubstantial differences from the literal languages of the claims.

What is claimed is:

1. A communication device for generating enhanced audio signals, the communication device comprising:

a first receiver receiving noise signals from a near-end location;

a second receiver configured to receive audio signals from far-end communication devices;

a processor configured to process the noise signals and audio signals to enhance the audio signals,

wherein firstly a mean noise power of the noise signal is estimated and secondly a linear predictive coding coefficients and gains are calculated based on the mean noise power to generate enhanced audio signals;

an earpiece to output the enhanced audio signals

wherein the gains comprise a first gain calculated within a first predefined range and a second gain calculated within a second predefined range.

2. The communication device of claim 1, wherein the first predefined range is between 10% to 30%.

3. The communication device of claim 1, wherein the second predefined range is between 70% to 90%.

4. The communication device of claim 1, wherein sum of the first gain and the second gain is 100%.

5. The communication device of claim 1 wherein the processor is configured to:

attenuate the audio signals with the first gain to generate a third audio signal;

attenuate the audio signals with the second gain to generate a fourth audio signal;

filter the fourth audio signal buffer by using LPC coefficients of the noise signals to generate a fifth audio signal;

add the fourth audio signal buffer to the fifth audio signal buffer to generate the enhanced audio signals outputted by the earpiece.

6. The communication device of claim 5, further comprising a memory configured to store the noise signals, the audio signals, the third audio signal, the fourth audio signal, the fifth audio signal and the enhanced audio signals and to store one or more program instructions executable by the processor.

7. The communication device of claim 1, wherein the processor is further configured to remove direct current components from the noise signals based on the mean noise power.

8. A method for generating enhanced audio signals, the method comprising the steps of:

receiving noise signals by a first receiver from a near-end location;

receiving audio signals by a second receiver from far-end communication devices;

configuring a processor for processing the noise signals and audio signals to enhance the audio signals,

wherein firstly a mean noise power of the noise signal is estimated and secondly a linear predictive coding coefficients and gains are calculated based on the mean noise power to generate enhanced audio signals;

outputting the enhanced audio signals using an earpiece;

wherein the gains comprise a first gain calculated within a first predefined range and a second gain calculated within a second predefined range.

9. The method of claim 8, wherein the first predefined range is between 10% to 30%.

10. The method of claim 8, wherein the second predefined range is between 70% to 90%.

11. The method of claim 8, wherein sum of the first gain and the second gain is 100%.

12. The method of claim 8 further comprising the steps of configuring the processor to:

attenuate the audio signals with the first gain to generate a third audio signal;

attenuate the audio signals with the second gain to generate a fourth audio signal;

filter the fourth audio signal buffer by using LPC coefficients of the noise signals to generate a fifth audio signal;

add the fourth audio signal buffer to the fifth audio signal buffer to generate the enhanced audio signals outputted by the earpiece.

13. The method of claim 12 further comprising configuring a memory to store the noise signals, the audio signals, the third audio signal, the fourth audio signal, the fifth audio signal and the enhanced audio signals and to store one or more program instructions executable by the processor.

14. The method of claim 8 further comprising configuring the processor to remove direct current components from the noise signals based on the mean noise power.

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